ABSTRACT OF THE DISCLOSURE

An apparatus and method for tuning voice playback ratio (pbr) to optimize call quality in a packet voice communications system, while taking into account network conditions. The pbr is the ratio of resampling rate to the original sampling rate. The invention also optimizes jitter buffer length (jb_0) for call quality. Between bursts of speech, the preferred embodiment of the invention optimizes call quality by varying the initial jb_0 and the pbr to achieve the best R-factor (R). R is an estimate of customer satisfaction with the quality of a voice call in real time. During bursts of speech, the value of jb_0 is fixed at the beginning of the BOS and the pbr is varied to achieve the best R . The method can be implemented during a burst of speech and between bursts of speech whenever the network conditions change.